Subject: Non-oversampled v. Oversampled
Posted by Wayne Parham on Sun, 11 Jan 2004 16:31:11 GMT
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Akhilesh Bajaj and I discussed this issue briefly last night, and I'd like to see the subject opened for discussion here. Both he and I are in the IT field, and quite familiar with the technologies required to implement either mechanism. But neither of us has really paid a great deal of attention to the number of actual algorithms chosen, other than to notice the debate and read about a couple of the most common types.Personally, I would have felt that using a higher oversampled rate would be best, and then to use interpolation between sampled words. That is my initial gut reaction, without giving it much thought. This is sort of like the simple and obvious idea that higher sampling rates and greater bit density always produce a higher quality and truer representation of the original signal.But oversampling an existing dataset isn't the same thing as recording with a higher sampling rate. Oversampling simply makes up some steps in between two samples, and fills in the blank. You can "make up" whatever you want to fill the gap, and this, I suppose, is what creative algorithm writers develop. The most obvious one is to use a simple interpolation scheme. But that can be as easily done with an electrical low-pass filter. Integrate the output of the DAC with a capacitor, a simple RC filter.This makes a great deal of sense, really. A simple one-to-one sample, output through a low-pass filter that smoothes the top end. An anaolog computer, after all, is like a very high-resolution, high speed digital computer. So the simple RC filter is an analog computer than does the interpolation very well.Of course, the trouble with this reasoning is that the 44.1kHz is very close to the minimum required to reproduce 20kHz. Each half cycle of a 22kHz will be sampled twice, and the DAC translation then comes closer to approximating a square wave. The RC filter would smooth the edges to approximate a sine, but then this requires a very high-order "brick wall" filter to attenuate the harmonics without attenuating the fundamental. First harmonic is only one octave away, and so filter slope must be high and the corner must be placed immediately above 20kHz.Another trouble with such a low sampling frequency is that of aliasing. Signals in the top-octave will usually not synchronize with the sampling frequency so that some of time, the output representation will skip cycles. If you sample a 20kHz sine signal with a 44kHz ADC, then you will develop a sort of beat-frequency representation of the signal. The first sample may pick up the leading 0.707 point of the first half cycle, and the second sample would then be just before the trailing 0.707 point. The third cycle would be between the falling zero crossing point and the ramp down towards the leading 0.707 point of the second half cycle. The fourth sample would be around the peak of the second half cycle. So the recorded dataset is aliased. Anti-aliasing must be the biggest challenge to CD algorithm developers, and not so much the relatively simple integration filter on the DAC's output.If data rate and storage isn't an issue, the best thing is to use a very high sampling rate and large word length. The higher, the better. You get to a point where filtering is not required and aliasing doesn't occur. But when the 44.1kHz rate is assumed, then massaging the recorded data with anti-aliasing and other digital processing will probably yield improvements to the output analog signal representation.What are your thoughts?

Subject: PAM implementations...
Posted by Magnus on Wed, 14 Jan 2004 21:15:07 GMT

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Wayne,it would indeed be very interesting so know what sort of PAM alghorithms most CD players use today! If it is simply a linear interpolation the low-pass filter solution would indeed produce similar results. After all, you need a low-pass filter after the DAC anyway.../Magnus

Subject: Re: Non-oversampled v. Oversampled
Posted by akhilesh on Thu, 15 Jan 2004 18:40:30 GMT
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HI Wayne,I am not an electrical engineer, so please take what i say with a barrel of salt. It seems to me that if we are philosophically opposed to creating information that is not there (interpolating), then a no-oversampling dac, with very high order filters (11-12-15-20?) may work.I suspect the reason commercial enterprises don;t sell these is cost. I may be wrong...there be something else to oversampling. On a qualitative note, I have talked to people who sell the inexpensive ($500-1000) non-oversampled DACs on the web, and to a person they either :a)refuse to take credit cards orb) entertain returns orc) sell kits that are non returnable. The goal may be to create a DAC with non oversampling, that will have a very high order filter, and also a great analog output stage.....hmmmmmdoes audio note do that?

Subject: Re: Non-oversampled v. Oversampled
Posted by Wayne Parham on Thu, 15 Jan 2004 20:47:36 GMT
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Personally, I can see the merit in oversampling. Naturally, I'd rather have the recording made with high resolution samples done at a high rate. But since the recording is already made, another good option is to oversample and interpolate in hi-res. I can see the merit in this approach.But for the tube guys, I can also see the attraction of a simple DAC that just spits out exactly what it gets. They roll of the top end with a simple filter function and let it fly. You're right that Audio Note makes some products like this. Maybe David Cope or Peter Qvortrup will comment at some point.

Subject: Re: Non-oversampled v. Oversampled
Posted by DRC on Fri, 16 Jan 2004 19:27:41 GMT
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Well, this will raise the hackles of the technically oriented, but not only do I find non-up/non-over sampled DACs to sound far more natural and musical than their "cleverer" counterparts, it gets worse. The filters used by nearly everyone really do a job on the musical value of the signal. Digital filters are the worst, but even analog filters do harm.Get all the "corrective" crap out of the way, and the music can come shining through. No, the 'scope shots won't look pretty. In fact, they'll look like hell. But the music, man oh man . . . . . .This is an opinion. It is based on extensive, but not exhaustive, listening. The listening includes upsampled, oversampled and non-oversampled DACs, the latter with and without analog filters which were initially chosen as the lesser of two evils. (The more evil being digital filters.)I won't argue the technical merits, but I can direct anyone who's interested to places where they can listen for themselves.

Subject: Re: Non-oversampled v. Oversampled
Posted by Wayne Parham on Fri, 16 Jan 2004 20:21:48 GMT
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Hi Dave!I can see merits in using non-oversampled DAC's. Just output exactly what the DAC gets, without interjecting anything in between samples. Any ultra-HF artifacts are filtered by the upper rolloff of the system. That's the filtering mechanism of a setup like this - It's simple and attractive for those going for a minimalist solution, which is important to many of those that like SET's, reduced component counts, small signal path, etc. So I think there are advantages that can be exploited in both oversampled and non-oversampled approaches.Wayne

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Subject: Re: Non-oversampled v. Oversampled
Posted by DRC on Tue, 20 Jan 2004 14:24:58 GMT
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Audio Note does nothing but non-oversampling DAC's, and we pride ourselves on great analog output stages, (transformer coupled rather than cap coupled from Level 2 Balanced on up), but, having started with analog filters rather than digital filters, we're moving as quickly as possible to NO filters. These choices and changes have all been based on audible musical outcome of each.The Level 4 DACs were built that way from the start, and Level 5, 3.1x Balanced and 2.1x Balanced are going that way as we speak. I'm waiting for the kit to upgrade my 3.1x Balanced here, and I can't wait to hear the results.

Subject: Re: Non-oversampled v. Oversampled
Posted by Wayne Parham on Wed, 17 Mar 2010 20:38:36 GMT
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I think another worthwhile discussion is PCM (fixed width Nyquist) verses Delta Sigma (one-bit oversampled bitstream) converters. It really is all about the conversion.